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Model Identification and Update Under Operation of Active Noise Control Systems

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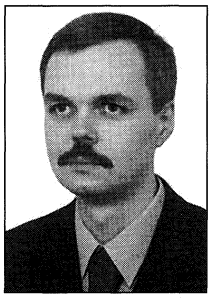

studia na Wydziale Automatyki, Elektroniki i Informatyki Politechniki Śląskiej w Gliwicach ukończył w roku 1986 na kierunku Elektronika, specjalności Automatyka Przemysłowa i Pomiary. W 1992 roku uzyskał stopień doktora nauk technicznych, a w 2000 roku stopień doktora habilitowanego nauk technicznych. Od ukończenia studiów jest pracownikiem Instytutu Automatyki Politechniki Śląskiej. Jego zainteresowania badawcze obejmują m.in. komputerowe systemy sterowania, identyfikację modeli matematycznych obiektów i sygnałów, sterowanie adaptacyjne oraz syntezę sygnałów o zadanych własnościach czasowych i widmowych.
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Streszczenie

W pracy zaprezentowano nowe spojrzenie na problem identyfikacji on-line modeli torów elektroakustycznych w adaptacyjnych kompensacyjnych układach aktywnego tłumienia hałasu (ATH). Problem ten dyskutowany jest jako problemem identyfikacji obiektu pracującego w układzie ze sprzężeniem zwrotnym przy niskim stosunku sygnału do szumu. Sprzężenie to wprowadzane jest w układach ATH poprzez algorytm adaptacji. Może być ono również wynikiem akustycznej interakcji głośnika sterującego z mikrofonem błędu. W literaturze poświęconej aktywnemu tłumieniu hałasu proponuje się, by w trakcie pracy układów ATH identyfikować modele torów elektroakustycznych wraz ze wspomnianym sprzężeniem zwrotnym. Otrzymane w ten sposób oceny parametrów modeli torów elektroakustycznych są obciążone. W artykule zaproponowano nowa metodę identyfikacji tych modeli. Metoda ta pozwala uzyskać asymptotycznie nieobciążone i zgodne oceny parametrów modeli torów elektroakustycznych.

Summary

In the paper, a new look at on-line electro-acoustic plant model identification and update for feedforward active noise control (ANC) systems under their operation is presented. The problem of on-line electro-acoustic plant model identification under active ANC system is discussed as a closed-loop identification problem with a low signal-to-noise ratio. The feedback is introduced by an adaptation algorithm. Additionally it may be also implied by an acoustic interaction between a control loudspeaker and reference microphone. In ANC literature concerning on-line electro-acoustic plant modelling it is proposed to identify electro-acoustic plant models together with the feedback. It leads to biased and inconsistent identification results. In the paper, an approach to this problem that gives results converging asymptotically to true electro-acoustic plant models is presented.

Key words: active noise control, system identification, adaptive systems, frequency response; Fast Fourier transform

1. Introduction

Active noise control (ANC) is concerned with attenuation of unwanted low frequency sound (noise) using electro-acoustical devices [16]. Its idea was originated in the 1930s [15]. However, recent research results in the fields of digital signal processing, control theory, system identification and development of digital signal processing hardware have made ANC a truly practical tool. Present-day ANC implementations are mainly done using digital adaptive feedforward or feedback systems.

To design an ANC system there is a need to identify models of electro-acoustic plants (secondary path and acoustic feedback path) before activating the ANC system [12]. These models should be also identified and updated on-line under operation of the ANC system because the electro-acoustic plants may be time varying. Additionally, performance of the

ANC system is affected by how well the electro-acoustic plants are identified. In the paper, a new look at on-line ANC system identification is presented. The problem of on-line electro-acoustic plants model identification with active adaptation algorithm is discussed as a closed-loop identification problem with a low signal-to-noise ratio. The feedback is introduced by an adaptation algorithm and may be also implied by an acoustic interaction of a control loudspeaker and reference microphone. The discussion is concentrated on on-line electro-acoustic plants model identification for feedforward ANC systems.

In the ANC literature existence of the feedback introduced by the adaptation algorithm and acoustic interaction of a control loudspeaker and reference microphone under on-line electro-acoustic plants model identification for feedforward ANC systems has not been noticed. It is proposed to identify models of electro-acoustic plants together with the feedback [10, 13, 20]. It leads to considerably biased and inconsistent results of model identification. In the paper, an approach to on-line electro-acoustic plants model identification for feedforward ANC systems is proposed. The proposed method uses external multisine excitation [2, 3, 7] added to the control signal. Estimates of electro-acoustic plant models are calculated using specially averaged control, error and reference signals. This averaging reduces efficiently influence of disturbances on identification results and identified models converges asymptotically to true electro-acoustic plant models even for low signal-to-noise ratio.

2. Feedforward ANC system

The block diagram of a typical adaptive feedforward ANC system [13] creating a local zone of quiet surrounding a single (error) microphone in a reverberant enclosure is shown in Fig. 1. The ANC system is working with the sampling interval T. The enclosure is disturbed by a noise (generated by a primary source), which should be reduced using a secondary source (control loud-speaker). It is assumed that the noise is a zero-mean random process with autocorrelation function vanishing to 0 for lags tending to infinity. A reference microphone placed near to the primary source is used to measure the reference signal $x_m(i)$. The disturbance path represents an acoustic space between the reference and error microphones. The secondary path is composed of D/A converter, reconstruction filter, amplifier, control loudspeaker and an acoustic space between the loudspeaker and error microphone. The acoustic wave generated by the control loudspeaker goes not only to the error microphone but also reaches the reference microphone. This interaction is called an acoustic feedback. The corresponding acoustic feedback path is composed of D/A converter, reconstruction filter, amplifier, control loud-speaker and the acoustic space between this loud-speaker and the reference microphone. The acoustic feedback is compensated by an additional filtration of the control signal $y(i)$ by a linear

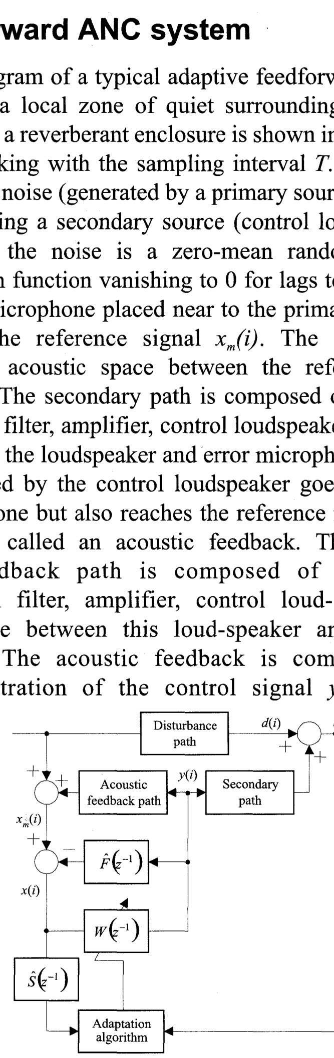

Fig 1. ANC system.

approximation of acoustic feedback path represented in Fig. 1 by the transfer function $\hat{F}(z^{-1})$. A digital linear filter is used as the compensator $W(z^{-1})$. Its coefficients are tuned on the basis of the error signal $e(i)$ and signal $x(i)$ filtered through a linear model of secondary path - the transfer function $\hat{S}(z^{-1})$. The goal of the adaptation algorithm is to calculate the coefficients of digital filter $W(z^{-1})$ that minimise the mean square value of error signal $e(i)$.

3. Identification problem

The structure of ANC system implies that there is a need to identify off-line models ($\hat{S}(z^{-1})$ and $\hat{F}(z^{-1})$) of electro-acoustic plants before activating the ANC system. These models have a great influence on performance of the ANC system [18]. Errors in their estimates may result in non-stability of the ANC system or may reduce adaptation algorithm speed of convergence. It is also worth to note that for some applications the electro-acoustic plants may be time varying. Its parameters may change due to movements of microphones, secondary source, people inside the enclosure or the air temperature and humidity variations. This implies that to assure convergence of the adaptation algorithm and stability of the ANC system the electro-acoustic plant models should be identified and updated when the ANC system is in operation. This will increase robustness of the ANC system. In the sequel, the discussion is concentrated on on-line secondary path model identification. Model of the acoustic feedback path may be identified on-line in the same way.

The problem of on-line secondary path model $\hat{S}(z^{-1})$ dentification (based on measurements of signals $y(i)$ and $e(i)$) with active adaptation algorithm is a closed-loop identification problem - the feedback is introduced by the adaptation algorithm and by an acoustic interaction of the control loudspeaker and reference microphone. It implies that the secondary path input $y(i)$ is correlated with the disturbance $d(i)$ at secondary path output. This is one of the most important reasons, why methods giving consistent estimates for plants without feedback (e.g. classical frequency analysis) may not work properly if they are used in a direct way to plants operating in closed-loop [8, 9, 11]. To avoid this problem external excitation signals are used in closed-loop identification methods [14, 17, 19].

Additional feature of on-line secondary path model identification is low signal-to-noise ratio because variance of the external excitation should be chosen so as not to decrease radically noise attenuation obtained by the operating ANC system. It is well known that in this case identified models of plants operated under closed-loop may be considerably biased and inconsistent [9, 17, 21]. To overcome this problem it is proposed in ANC literature to identify the secondary path model using measurements of signals $u(i)$ and $e(i)$ [10, 13, 20]. Of course this procedure also results in biased estimates because the secondary path is identified together with the mentioned feedback. The above problem is illustrated by the following example. In this example an operating ANC system is modelled by its linear approximation. The corresponding block diagram is presented in Fig. 2. In this figure $\hat{S}(z^{-1})$ the transfer function of secondary path, $C(z^{-1})$ is a transfer function of the remaining part of ANC system seen from the input and output of secondary path. $u(i)$ is the additional external excitation signal.

The secondary path frequency response estimate $\hat{S}(j\omega T)$ for frequencies $\omega T \in [0,2\pi)$ may be calculated as the ratio of two frequency response estimators of plants operating without feedback: the first one $\hat{K}_{eu}(j\omega T)$ between $e(i)$ and $u(i)$ to the second one ($\hat{K}_{yu}(j\omega T)$) between $y(i)$ and $u(i)$:

$$\hat{S}(j\omega T)=\frac{\hat{K}_{eu}(j\omega T)}{\hat{K}_{yu}(j\omega T)}=\frac{S(j\omega T)\gamma(j\omega T)+1}{\gamma(j\omega T)-C(j\omega T)} \quad (1)$$

where $\gamma(j\omega T)=\dfrac{\Phi_{uu}(\omega T)}{\Phi_{dd}(\omega T)}$ is the signal-to-noise ratio; $\Phi_{uu}(\omega T)$ and $\Phi_{dd}(\omega T)$ are power spectral densities of $u(i)$ and $d(i)$, respectively. The above estimator is biased. Its biasedness declines with the increase of $\gamma(j\omega T)$. This increase may be obtained by increasing variance of the external excitation signal. However, it should be remembered that any increase of this variance implies lower noise attenuation. Additionally, the increase of variance is limited by the range of D/A converter. Increase of $\gamma(j\omega T)$ without increasing the variance of external excitation may be obtained by applying an identification method using a multisine excitation and a special method of data processing [2, 5]. It gives identification results that converge asymptotically to true secondary path model even for low signal-to-noise ratio. The block diagram of ANC system with on-line electro-acoustic plants model identification is presented in Fig. 3.

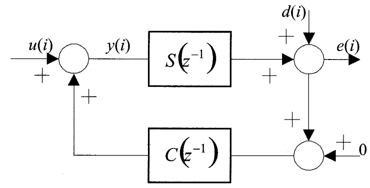

Fig. 2. A model of ANC system.

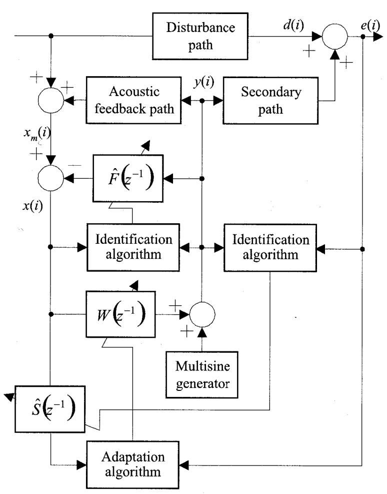

Fig. 3. ANC system with on-line electro-acoustic plants model identification.

4. Electro-acoustic plants model identification

The N-sample real-valued scalar multisine excitation [3, 4, 6] $u(i)$ is defined in the time-domain by a sum of $(N/2)+1$ discrete-time harmonic sines, including a constant component:

$$u(i)=\sum_{n=0}^{N/2} A_n \sin(\Omega n i+\varphi_n), \quad (2)$$

where i=0,1,...,N-1 denotes consecutive discrete time instants, $\Omega = 2\pi/N$ denotes the fundamental relative frequency, n=0,1,...,$N/2$ denotes consecutive harmonics of this frequency in the range $[0, \pi]$, A_n are deterministic amplitudes of the sine components and φ_n are the corresponding phase shifts, of which $\varphi_0 = \varphi_{n/2} = \pi/2$. The N-sample multisine excitation $u(i)$ may be effectively synthesised using Fast Fourier Transform algorithms.

The identification experiment with multisine excitation is designed in a special way. During identification experiment the secondary path is excited by a periodically repeated N-sample multisine excitation $u(i)$ that is added to the control signal. Start of the data acquisition is delayed to the time instant of putting the external excitation into the ANC system so as all transients caused by inputting external excitation into the system become extinct. Under these conditions the excitation is repeated m times.

In the proposed identification method model of the secondary path is identified on the basis of measurements of signals $y(i)$ and $e(i)$ obtained under operation of the ANC system. The model of the secondary path is identified on the basis of mN-sample secondary path input data sequence $\{y(0), y(1),..., y(mN-1)\}$ and the corresponding mN-sample output data sequence $\{e(0), e(1),..., e(mN-1)\}$. It is worth to mention that to estimate secondary path model measurements of external multisine excitation are not used.

The method of data processing used in the paper is based on the averaged $y(i)$ for time instants $i=0,1,...,N-1$ values of secondary path input signal [2, 7]:

$$\bar{y}(i) = \frac{1}{m}\sum_{s=0}^{m-1} y(i+sN) \qquad (3)$$

and obtained in the same way averaged $\bar{e}(i)$ values of secondary path output signal. The averaged N-sample data sequences $\bar{y}(i)$ and $\bar{e}(i)$ $(i=0,1,...,N-1)$ may be calculated recursively during identification experiment using an algorithm of on-line mean value calculation.

It follows from statistical properties of the noise that the averaged signals $\bar{y}(i)$ and $\bar{e}(i)$ are unbiased and consistent estimators of the noise free steady state secondary path input and output responses for one period of the external multisine excitation. Their variances decline with increase of the number m of processed data segments.

Secondary path frequency response estimates $\hat{S}(j\Omega n)$ for frequencies Ωn $(n=0,1,...,N-1)$ may be calculated using the empirical transfer estimator [7]:

$$\hat{S}(j\Omega n) = \frac{\bar{E}(j\Omega n)}{\bar{Y}(j\Omega n)}, \qquad (4)$$

where: $\bar{Y}(j\Omega n)$ is the N-point discrete Fourier transform of $y(i)$:

$$\bar{Y}(j\Omega n) = \sum_{i=0}^{N-1} \bar{y}(i)\, e^{-j\Omega ni} \qquad (5)$$

$\bar{E}(j\Omega n)$ is the corresponding N-point discrete Fourier transform of $\bar{e}(i)$.

The secondary path frequency response estimator (4) may be obtained as a ratio of two frequency response estimators of plants operating without feedback: the first one between $e(i)$ and $u(i)$ to the second one between $y(i)$ and $u(i)$. These estimators are unbiased estimators of the corresponding true frequency response for frequencies Ωn $(n=0,1,...,N-1)$. Their variances decline with the increase of m. The unbiasedness property is not inherited by the estimator (4) - this estimator is asymptotically unbiased. Its bias tends to 0 for $m \to \infty$. It is obvious that variance of obtained secondary path frequency response estimates also declines to 0 for $m \to \infty$. These properties of the proposed secondary path model identification method are not signal-to-noise ratio dependent.

The identified frequency response may be used directly to update secondary path model in adaptation algorithms using block processing technique [1] or may be recalculated into a FIR filter using the inverse discrete Fourier transform. It is also worth to mention that the averaged data sequences $\bar{y}(i)$ and $\bar{e}(i)$ may be applied to identify on-line parametric models of the secondary path using recursive least squares, instrumental variable or LMS identification methods. The acoustic feedback path may be identified on-line in the same way using averaged signals $\bar{y}(i)$ and $\bar{x}(i)$. Calculation of electro-acoustic plants model estimates and its update may be done periodically, for example after each repetition of N-sample multisine excitation.

5. Example

The error microphone was placed into a laboratory enclosure of about 23 m^3 of cubature. This enclosure was disturbed by a wide-sense zero-mean stationary gaussian random noise with the power spectral density presented in Fig. 4.

The control loudspeaker was placed at about 0.6 m apart from the error microphone. A reference microphone placed near to the primary noise source was used to measure the reference signal $x_m(i)$. During laboratory experiments the ANC system was working with the sampling interval $T = 0.002$ s. The compensator $W(z^{-1})$ was a FIR filter with 300 coefficients. These coefficients were adapted using LMS algorithm. Secondary and acoustic feedback paths were also modelled by FIR filters with 250 coefficients each. Their parameters were identified without acoustic disturbance before activating the ANC system. After switching the ANC system on the noise attenuation reached 5.9 dB. To identify the secondary path a periodically repeated 1024-sample white multisine excitation $u(i)$ with variance 0.36 V^2 was added to the output of filter $W(z^{-1})$ during operation of the ANC system. It caused that the noise attenuation decreased to 5.7 dB. Estimates of the secondary path frequency response $S(j\omega T)$ were calculated based on m=200 data segments using: a) the proposed approach and b) a FIR model of the secondary path (with 250 coefficients) estimated using LMS algorithm. In Fig. 5 magnitudes of obtained frequency response estimates are compared with a magnitude of frequency response identified without acoustic disturbance and with the ANC system off. A bias in the frequency response estimate obtained from estimated FIR secondary path model can be observed.

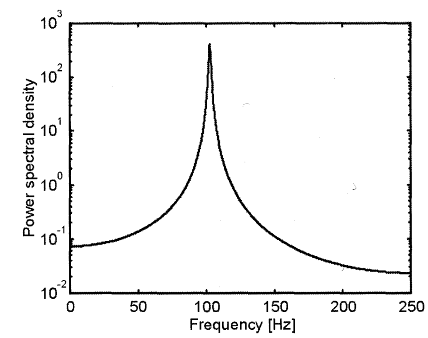

Fig. 4. Power spectral density of an acoustic disturbance.

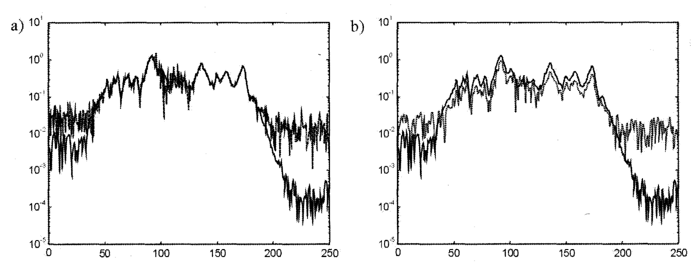

Fig. 5. Frequency response magnitude estimates of secondary path: ANC system off (solid line), ANC system on (dotted line) - the proposed approach (a) and LMS algorithm (b).

6. Conclusions

A problem of on-line electro-acoustic plants (secondary path and acoustic feedback path) model identification and update under operation of ANC system was discussed as a closed-loop identification problem with a low signal-to-noise ratio. An identification method that gives results converging asymptotically to true electro-acoustic plant models was proposed. The method uses external N-sample multisine excitation. During identification experiment the excitation is periodically repeated. One period of steady-state undisturbed secondary path as well as acoustic feedback path input and the corresponding output signals are estimated using time-domain averaging. This averaging reduces efficiently an influence of disturbances on identification results. Models of electro-acoustic plants may be calculated and updated periodically, for example after each repetition of the external N-sample multisine excitation.

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Tytuł: Identyfikacja i uaktualnianie modeli w trakcie pracy układów aktywnego tłumienia hałasu

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